elysia
ALPHA COMPRESSOR
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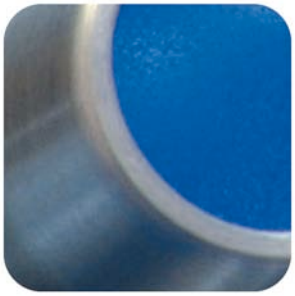

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Dear friend of audio culture,

First of all, we would like to thank you sincerely for choosing the alpha compressor as your new mastering tool. From now on, you not only have the maximum possible audio quality in dynamics processing at your disposal, but also a complete arsenal of flexible and effective tools which will shift the results of your work to a new level.

In short: You have opted for a product designed without the slightest compromise!

Please take a little time to read this manual thoroughly, as it will help you to entirely understand the enormous potentials and to really push the envelope. We have attached great importance to practical experience and fast results, which is also the reason for reserving the explanation of the technological excesses realized in the alpha compressor to our website.

The "Basics" chapter contains essential information for the fundamental comprehension and quick use of the specific functions and modules. After this, a couple of "Scenarios" are introduced in order to illuminate just some of the versatile applications of your new favorite compressor. If you want to go into further details, please have a look at the "Reference" chapter which elaborates on every particular parameter.

If you have further questions or comments, please do not hesitate to contact us – we enjoy being of your service. But for now, it's time to wish you lots of fun and pure audio pleasure with your alpha compressor.

Use the Force…

the elysians

Mounting

The best way to place the alpha compressor is to put it on top of a rack. In this position, ventilation will be optimal and the unit will not heat up too much.

In case you mount it into a 19" rack or vertically into a console, please leave a free space of about 2cm on top and bottom so that the air can circulate freely. Also make sure that the rack rails can really carry the heavy weight: the device weighs about 16kg that should be reinforced from the back in case of doubt. If required, the four aluminum stands can be unscrewed from the outside of the housing.

Dependent on your personal studio ambience, you might find the light intensity of the controllers and buttons too bright or too dark. This is no problem, because two separate wheel controllers on the back panel help you to adapt the lights to your particular needs.

Audio Path

Definitely no secret, but still worth mentioning: Inferior cables can noticeably decrease the entire audio quality; therefore no compromises should be made in this regard. All audio cables should be of a high quality and as short as possible.

Furthermore, only the processors that are needed every day should be constant parts of the signal chain. Not all compressors, equalizers etc. feature real hard bypasses; resulting in irretrievable losses of sonic details. The motto ‚less is more' is pretty obvious in this case.

Note: If a noticeable change in level occurs when the compressor is activated, balanced connection might be the reason. Please look at page 36 for further information.

Warm Up

When the compressor is powered up, it might happen that the Gain reduction meters fully light up even when there is no audio signal present (depending on the operation mode and the setting of parameters, both meters might even display different values). After a warm up time of about 15 minutes this will be gone automatically.

All temperature-sensitive circuits will be heated up by then and the compressor is ready for operation. Only now constant control characteristics and best possible audio quality can be achieved, therefore the warm-up time must be adhered to before every production.

Safety Instructions

Voltage: Please check if the voltage needed in your country is set correctly at the back side of the unit before powering it up.

Service: Make sure to have your unit serviced by authorized personnel only and contact elysia for technical support. Please open the alpha compressor only in case you really know what has to be done.

Environment: Avoid heat, cold, humidity, vibrations, dust and electronic (e.g. transformers) as well as digital (digital processors) stray fields.

Cleaning: Clean the unit with a dry, clean cloth. Never use chemical dissolvers or cleansers!

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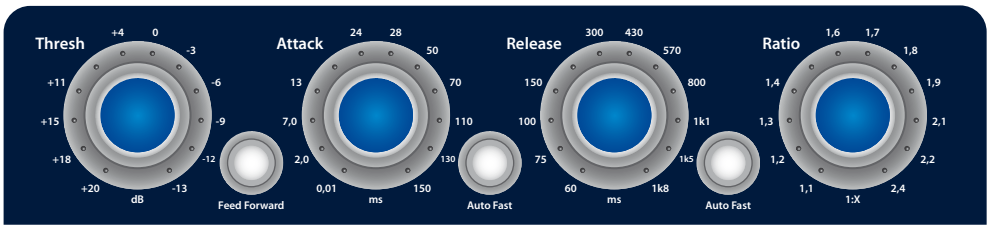
BASICS - Basic Design

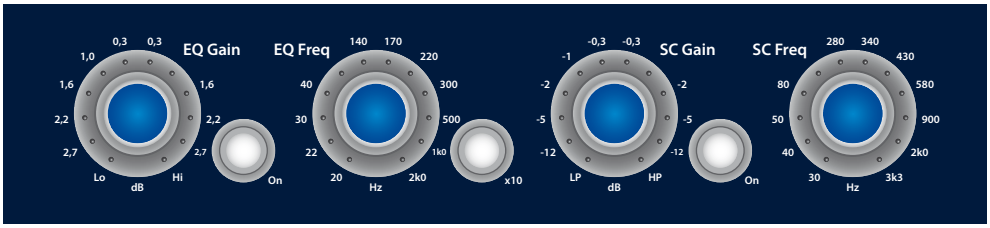
Never fear! With its many options and quite a lot of controllers the alpha compressor might look a little tricky at first sight. In practical experience, however, it is a clearly structured and easy to use tool.

The control elements are divided into three layers, whereas the left and the right channel sides are completely identical.



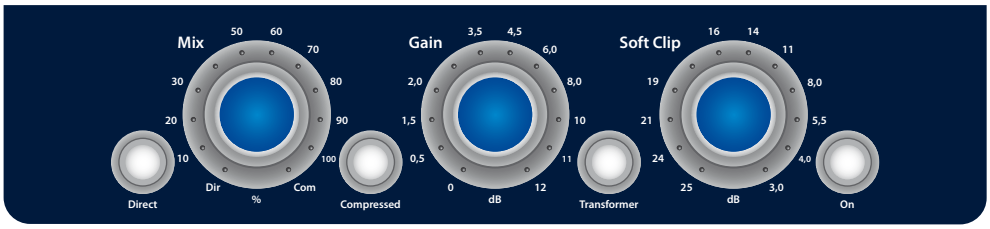
Compressor section

The upper array contains the dynamic section with the classic parameters Threshold, Attack, Release and Ratio. Furthermore, there are some interesting special functions like Feed Forward and Auto Fast that will be explained later.



Filter section

Firstly, the middle array features the Audio filter, which enables you to perform subtle changes on the overall tonal character. Secondly, it consists of the Sidechain filter which allows frequency-dependent compression when it is switched into the detector path.



Level section

The lower array starts with the Mix controller that allows blending between the unprocessed and the compressed, filtered and amplified signals. Next in line is the Gain controller to make up the gain of the reduced signal. The last one is the Soft Clip limiter which allows catching fast transients.

In the middle of the unit, right below the Gain reduction meters, there are three buttons to set the distinct operating modes of the alpha compressor:

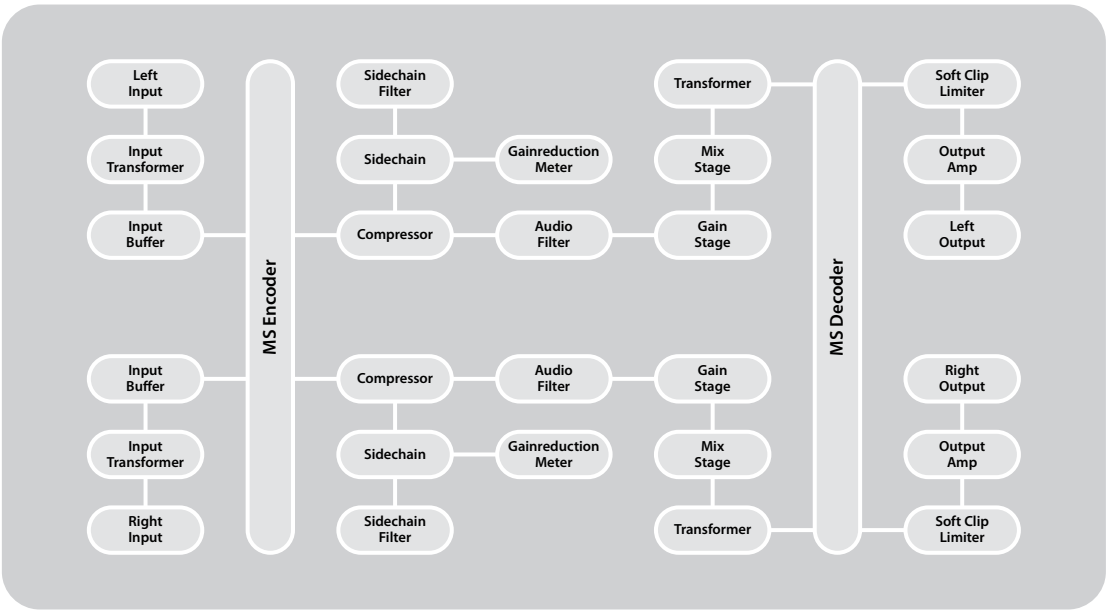
Active
If this button is pressed (LED glows), the incoming signals will be processed in the alpha compressor. Otherwise there will be a hard bypass, whereas the Gain reduction meters will still stay active.

MS Mode
If this button is pressed, the alpha compressor will work in MS mode in which the controls on the left side affect the middle channel and those on the right side belong to the side channel. Otherwise the unit will work in Stereo mode. The Soft Clip limiter will not be influenced by this setting as it always works in Stereo mode.

Channel Link
If this button is pressed, the left upper array will act as a master for the dynamic sections of both channels. All other parameters (Filter, Mix, Gain etc.) will not be linked, though, and therefore have to be set manually. In this mode the Gain reduction will be identical for both channels.

Block Diagram

The schematic shows the signal flow from the input through the specific modules to the output. MS matrix, Sidechain filters, Audio filters, Mix stages, Transformers and Soft Clip limiters can be optionally switched into the signal path via relays, which can also be done with the Auto Fast and Feed Forward functions.



Stereo Compression

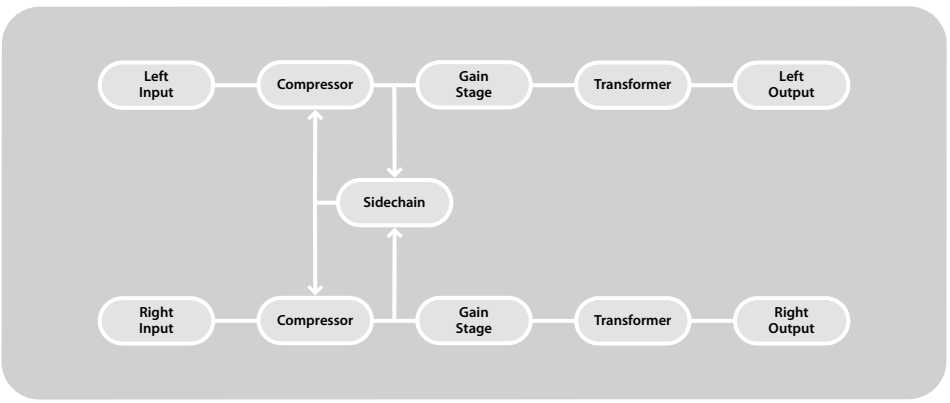
In order to get a first impression of the sonic quality and the control characteristics of the alpha compressor, only the controllers highlighted in the figure will be engaged. The functions that are currently not needed should be deactivated (related LEDs do not glow). Et voilà: We have a classic linked stereo compressor!

A track with a broad dynamic range is the best material for a first test. Just start playing with the Threshold, Attack, Release and Ratio parameters of the left channel in order to get to know the control behavior of the compressor. Allow yourself to take some time for this, as the understanding of the basic control characteristics is the most important condition for perfect results.

Keep having a look at the Gain reduction meters from time to time. No doubt: the ear is the decisive benchmark. Just consider the optical information as a supporting means to evaluate the control mode (fast/slow, strong/weak, etc.).

The level of the compressed signals can be made up by adding an appropriate amount of amplification with the Gain controllers. Since the Link function will only affect the control mode of the compressor, the Gain controllers of both channels should be set to the same values in Stereo Link mode. For instant AB comparisons just push the Active button.

Note: If the Threshold controller produces too little Gain reduction when turned hard right or too much of it when turned hard left, please go to page 28 and see how to adjust the internal Threshold.



Auto Fast (Attack)

The Auto Fast function causes an automatic shortening of the control time on fast and loud signal peaks. This effect is easy to comprehend when compressing a preferably dynamic drum loop, for example.

Please set the following values as a starting position:

- Attack: 50ms
- Release: 300ms
- Ratio: 1:1,8
- Threshold: Set to a position that results in approx. 3-4dB Gain reduction

Because of the quite long Attack time the snare and bass drum will be processed very moderately. If Auto Fast is now engaged, the compressor will react to the fast and loud peaks, resulting in a noticeably higher amount of Gain reduction.

And here is the trick: Directly after this process the Attack time will return to the value that was originally set, and on ‚slower' signals the original value will not be changed at all. The compressor will only become very fast in case it is really needed!

Auto Fast (Release)

The same example is also suitable to use the Auto Fast function for the Release parameter, too.

The following values are useful to get started:

- Attack: 30ms + Auto Fast
- Release: 300ms
- Ratio: 1:1,8
- Threshold: Set to a position that results in approx. 7-8dB Gain reduction

Now press the Auto Fast button of the Release controller and keep your ears open as well as an eye on the meter: the area between 4-8dB is reduced very fast now, whereas the set value of 300ms stays valid between 0 and 4dB.

Note: If you use very long Release times, the effectiveness of this function will decrease. Very long Attack times will also reduce the intensity of this circuit.

This function can be a great help especially for applications with difficult dynamic structures, as it counteracts the danger of clippings caused by too short values and the loss of loudness and pressure caused by too slow settings at the same time.

Feed Forward

This function makes it possible to switch the junction of the Sidechain alternatively behind (Feedback) or in front (Feed Forward) of the actual compressor section. This has an enormous effect on the character of the compressor.

Again, our drum loop can be used for demonstration: While processing in Feedback mode will run very smooth and even, switching into Feed Forward mode will result in a clearly more aggressive and harder kind of compression.

From the technical point of view this function mainly influences the characteristic curve of the Ratio value. In Feedback mode it goes up to a moderate Ratio of 1:2,5. In contrast, the Feed Forward mode provides much higher values that also allow limiter settings and even negative ratios (i.e. loud signals will be reduced even more). Thus, small changes in dynamics can generate a high amount of Gain reduction.

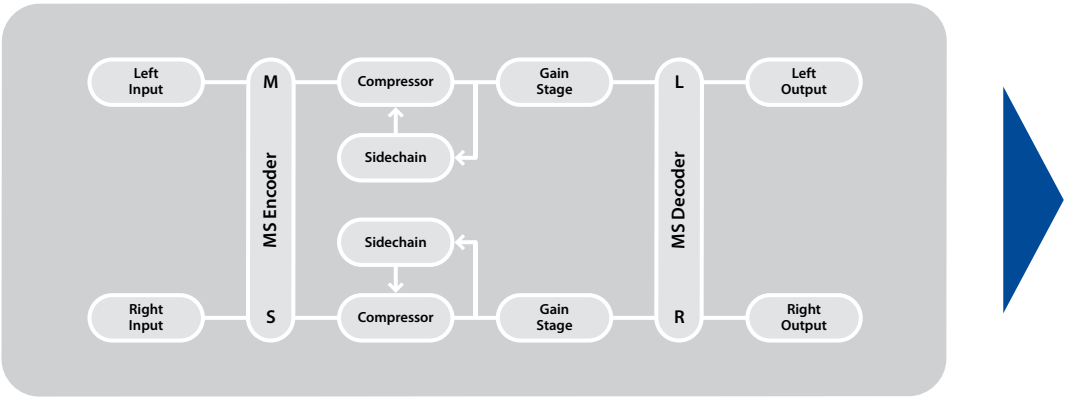
The values of the Attack controller will also change noticeably in Feed Forward mode, as they will become almost twice as high as the value shown on the scale.

Transformer

Each of the two channels features an additional transformer that can be switched into the signal chain after the Mix stage. In principle, these are classic output transformers, but here they are not used for balancing and galvanic isolation, but as an additional means of sound design.

If you like to add that certain amount of‚iron' to your sound, just push the Transformer button and there you are. Because of the mastering approach of the alpha compressor this feature is much more a subtle audio shaping feature than a glaring sound effect.

In Stereo mode the buttons for both channels should be activated or deactivated at the same time. Depending on the source material and personal taste, you can, however, also add the transformer sound to single channels when working in MS mode.



MS Matrix

One of the most powerful features of the alpha compressor is its switchable MS matrix. By activating it, the incoming stereo signal is encoded into a middle channel (mono) and a side channel which contains the stereo parts of the left and right channel (but not the common mono part).

This enables you to process the middle and the side signals separately, and finally both channels will be decoded back to left and right (stereo). A practical example easily helps to understand the general idea.

The ideal song for this would be the following: Important tracks like voice, drums and bass have been placed in the middle whereas the remaining instruments are mixed stereophonically.

Switch the alpha compressor into MS mode and leave the Link function deactivated for now. The left side of the compressor now controls the middle channel and the right side the side channel. By using the Compressed switches, you can listen to both channels separately, so that you will be able to hear what is instantly happening in the particular channel.

Now you can process the mono and side parts separately from each other. Depending on the mix, sometimes it is only the middle channel that needs compression whereas the side channel stays unprocessed. Other mixes have a lot of stereo information; in these cases both middle and side channels are processed, the settings of which can be completely different from each other.

At the beginning, both Gain controllers should be set at the same values, but that can change in the progress of optimization by all means. You can use different settings of the Gain controllers in MS mode for specific changes of the stereo width, for example.

Of course the Link function also works in MS mode. In this case, signals in the M and S channels are reduced evenly so that the result of the compression is the same as if working in stereo mode. However, you will still be able to adjust the level proportion between M and S with the Gain controllers.

Sidechain Filter

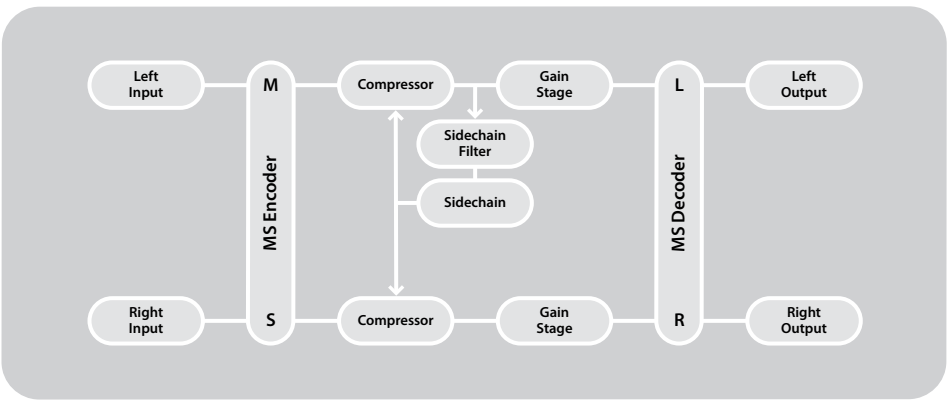
The Sidechain filter allows frequency-dependent shaping of the compression by giving specific frequency areas a stronger or weaker influence on the detection circuit.

The linked MS mode is most suitable to demonstrate the effect: Since most signal energy can be found in the middle as a general rule (bass drum etc.), the filter will have its greatest impact here. A track with a balanced proportion of bass, middle and treble frequencies should be chosen to give this function a try.

If the SC Gain controller is set to HP (High Pass), the filter will act like a 6dB high pass and the reaction of the compressor on bass frequencies decreases. The setting LP (Low Pass) turns the filter into a 6dB low pass and the compressor reacts primarily on low frequencies. Again, the Gain reduction meter is a reliable aid to evaluate the distinct modes of operation.

The combination of Sidechain filters, MS matrix and different Attack settings enables you to make very selective changes and – depending on the source material – even to process single instruments or voices in finished mixes. Details on that can be found in the "Scenarios" chapter.

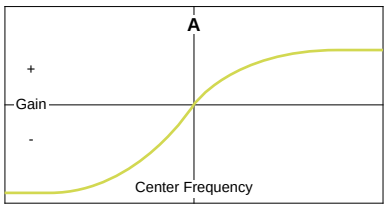
Note: When working in linked Stereo mode, the Sidechain filters of both channels should be set to identical values. In linked MS mode, however, only the Sidechain filter of the middle channel is active, and in unlinked MS mode the SC filter of the side channel is available, too.

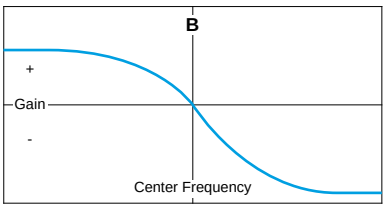


Audio Filter

The Audio filter is placed behind the compressor, avoiding unwanted influences on the detection circuit of the dynamic section.

If the EQ Gain controller is turned hard right, the signal rates above the selected center frequency will be boosted by 3dB and the part below the center frequency will be cut by 5dB [Fig. A]. Turning the controller hard left reverses the effect: Below the center frequency there is a boost and above it there is a cut [Fig. B].

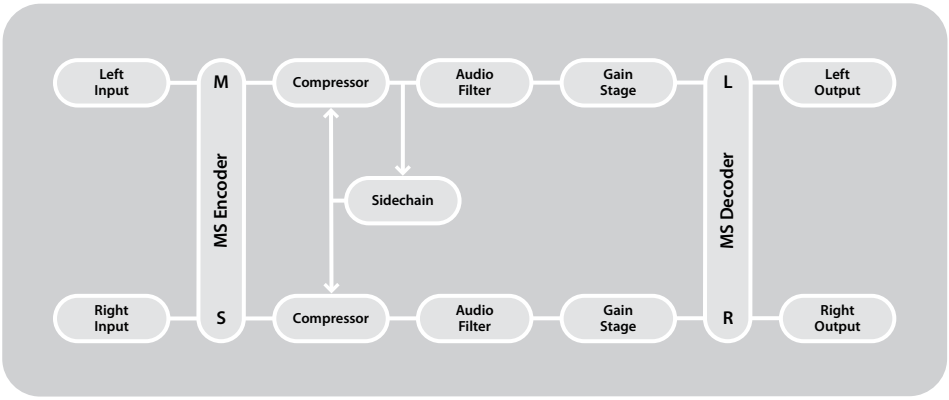




The EQ Freq controller comprehends an array from 20Hz - 2kHz which can be shifted to 200Hz - 20kHz by pushing the x10 button.

In the linked MS mode shown above it is possible to make different settings for the Audio filters in the M and the S band. Again, the results can be listened to separately by using the Compressed switches for each channel. However, in Stereo mode, both Audio filters should be set to the same values in order to avoid unwanted shifts in the stereo spectrum.

Note: Depending on the settings of the Audio filters it might be necessary to adapt the volume with the Gain controllers.



Mix Controller

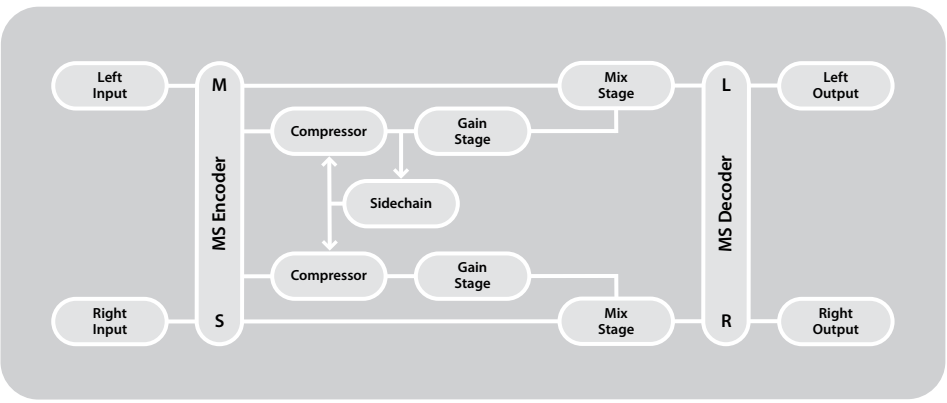
The Mix controller makes it possible to cross-fade between the unprocessed and the compressed/filtered signals. This allows parallel compression right in the alpha compressor; and renders additional routings in favor of a better signal quality unnecessary.

Now you can use even extreme compression settings without killing the track by winning the loudness war. By mixing just a part of the compressed signal to the original one the major portion of the initial dynamic structure remains.

The control logic of this feature is also worth mentioning. If only the Compressed button is pushed, you will hear the compressed signal only. Otherwise, if only the Direct button is pushed, only the unprocessed signal routed from before the compressor section will be audible. If both buttons are pushed, the Mix controller will become active, and if none of the buttons is pushed, the channel will be muted.

In practice you can switch between unprocessed, compressed or mixed signals very fast without having to change the position of the Mix controller.

In this way, the left and right channels (Stereo mode) respectively the middle and side channels (MS mode) can be listened to separately – here you have the choice again between the original, the compressed or the mixed signal. To mute the other channel, just deactivate the associated Direct and Compressed buttons.



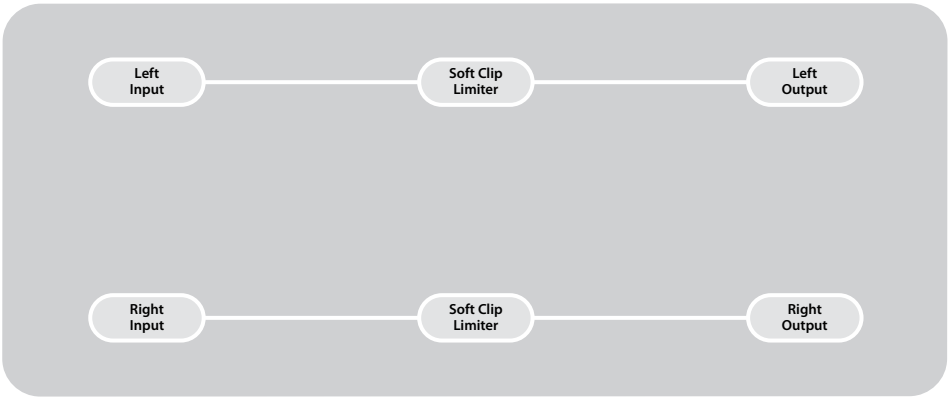
Soft Clip Limiter

The Soft Clipper was designed to limit short, loud transients, and thus protecting subsequent AD converters from clipping. However, the circuit does not behave like a classic brick wall limiter, but more like an analog tape machine driven into saturation: it rounds the peaks instead of making hard cuts.

The Soft Clip modules are placed directly in front of the output stages. Their settings are not influenced by the Mix and Gain controllers. This is also the case in MS mode, because the limiters are located after the MS decoder and therefore are always assigned to the left and right stereo channels. Note that the channel link does not affect the circuit either – because of this the settings of the Soft Clip controllers for both channels should always be identical!

If the alpha compressor is the last unit in the signal path before the AD converter, it is only once that the setting of the Soft Clip limiter has to be adapted to the converter. The best would be to use a fast, sample-accurate PPM meter to find the right setting, because it is an accurate method of benchmarking the resulting decrease in level.

The Soft Clip LEDs located on top of the Gain reduction meters should only light up shortly, as noticeable distortion appears if you drive this circuit too hard. Particularly with regard to acoustic and classical music, this feature should be used with care and moderate settings only.



SCENARIOS - Stereo Linked

This setting is especially useful for tracks in which the instruments are not placed straight in the middle of the stereo spectrum – when a recording has been made with the use of classic stereo miking like in Jazz, for example.

In order to come up to such styles of music you should make sure to create an unobtrusive and smooth kind of compression. This can be achieved by setting a long Attack time and a Ratio value in the lower control range, at which only a very few dB of Gain reduction are required.

If necessary, you can use the Mix controllers to apply the effect of the compressor in even finer doses (both Direct and Compressed buttons have to be activated for this).

Moreover, you can kick in the Audio filters to make additional sonic corrections.

Note: Look after setting the Mix, Gain and Filter controllers at the same values when working in Stereo mode. The Link function only affects the control mode of the actual compressor section.

Of course the linked Stereo mode can also be used for many other styles of music, but often you'll be able to achieve even better results in MS mode.

MS Linked

The function of the compressor in linked MS mode is quite similar to its function in linked Stereo mode. The time parameters are set with the controllers on the left side of the unit, and the resulting Gain reduction in the middle and the side channel will be exactly the same.

The figured settings are a good starting position for many pop and rock tracks: the Attack time of 30-40ms ensures that transients from percussion instruments will hardly be compressed, while the Release time of 250ms lets the compressor decrease its Gain reduction fast enough.

Variations of the Release time are especially effective in controlling the attained amount of loudness. Here, activating the Auto Fast function should be preferred to setting a very fast Release time.

The generated Gain reduction is primarily affected by matching the Threshold and Ratio parameters. A Ratio of 1:1,7 has often proved to be a good start.

But what is the essential advantage of the linked MS mode compared to the linked Stereo mode? The levels of M and S can be set differently, which makes it very easy to perform changes of the stereo width (Mix too narrow? More gain to the side channel. Not enough punch? Add gain to the middle). Furthermore, a left/right drift caused by possible unequal settings of the Gain controllers is impossible in this mode.

MS Unlinked

The true strengths of an MS compressor will not be revealed until it is switched into unlinked MS mode.

Depending on the mix and the style of music the dynamics and loudness in the middle and side channel can be completely different from each other. Take a heavily pumping bass drum right in the middle and 100% stable strings in the sides, for example – just try this with any other compressor ;-)

Thus it can make perfect sense to set strongly differing values of Threshold, Attack, Release and Ratio for the middle and the sides. In order to achieve a better evaluation of the results, it is advisable to listen to the processed channel solo for a start.

Please remember to deactivate the Direct and Compressed buttons of the other channel for this.

Using the Audio filters, which can also be set completely different from each other now, can help to additionally enhance the tonal balance of a mix.

Once you have discovered the potentials of the MS mode, lots of formerly difficult mixes can be controlled much better, resulting in audibly better effects.

Upward Leveling

Upward Leveling – this describes the possibility to make only the low signal parts louder and leave the louder changes in dynamics unprocessed.

The settings shown above are very suitable for classical music, for example, where ‚typical' compression is objectionable. The compressor works in linked Stereo mode, the Attack time is set at maximum and the Release time is at 150ms. Under these conditions the alpha compressor will function as a leveler.

In Feed Forward mode the Attack time is twice as long compared to Feedback mode and therefore amounts to 300ms. The Ratio characteristic corresponds to the one of a limiter, and the make up gain is about 6dB.

The result of these settings: Quieter passages that do not produce any Gain reduction will be boosted by 6dB, whereas loud signals will be limited by the compressor.

Now the special trick is to set the Mix controller to 50%. By adding the original signal the initial dynamics will be saved, and the bottom line is that only the quieter signals will be turned up.

MS Leveling

As with Upward Leveling, long Attack times are the secret of MS Leveling.

Here the compressor reacts primarily to the complete signal energy and less to single rhythmic accents. If this principle is now used on the side and the middle channel separately, a stereo mix will become more compact in an unobtrusive way and everything will sound a little more vivid.

This method is especially interesting to enhance rock and pop tracks in a discreet but effective way.

Here the Feed Forward mode is also active. The Ratio is set at about 1:1,5 and the Gain controller at about 3dB. The Threshold is adapted to a value that results in approx. 3-4dB.

Groove Compression

Sometimes it would be great to be able to just emphasize the rhythmical elements of a mix in order to bring out the groove and therefore giving the track the right kick.

The figure shows how the alpha compressor can help you to cope with this challenge. The bass drum is used as a trigger for the compressor; therefore the Sidechain filter is of special importance in this application. It is set to Low Pass, with a center frequency of approx. 70Hz.

Now the Threshold and Ratio controllers are used to set the intensity of the Gain reduction, and the groove will be controlled by the Release parameter, which has merely to be adapted the speed of the track.

Depending on the song and the style of music the Gain reduction can be higher by all means; usually 5-6dB will do very good.

The linked MS mode makes sense for this application, because the influence of the bass drum on the control mode is enhanced as well as the resulting sonic effect.

Vocal Down

If you have a track in which the vocals are too loud, the compressor can be set in a way that it mainly reacts to the lead vocals placed in the middle, resulting in a better integration into the mix.

For this purpose switch it into the unlinked MS mode. Again, the specialty here lies in combining the Sidechain filters and the Attack parameters. The SC filter of the middle channel is set to high pass at a center frequency of about 270Hz, the effect of which is that the compressor hardly reacts to bass impulses anymore.

The Attack time lies between 70-100ms. Thus, transient-like signals like a snare drum will also only have a minor influence on the control mode.

The Release time is set at about 100ms so that the compressor can return to its idle status fast enough.

In the side band both the controller for the Threshold and the Ratio parameters should be turned hard left if you do not wish any further processing there. An equal setting of the middle and side channel Gain controllers is also advised.

DeEssing

A variation of the Vocal Down scenario is DeEssing which is used for reducing unwanted ess and hiss sounds.

The Sidechain filter of the middle channel is set to approx. 2kHz, which is even higher than in the previous application, so that only the high frequencies can influence the detection circuit.

At about 100ms the Attack time is also quite long and the Release time is set to a fast 50ms. Now Threshold and Ratio are used to set the trigger point and the intensity of the processing.

This application runs in unlinked MS mode again to assure that only the middle channel is processed (but of course you can still process the sides separately if needed).

Parallel Compression

If it should (or must) become really loud, you can also use the alpha compressor to drive a track against the wall – but without the negative effects this usually entails.

For this procedure it is preferable to work in the classic linked Stereo mode.

The time parameters are set very fast here; especially the Attack time at only 10µS. The Release parameter is set at 150ms while the corresponding Auto Fast function is also activated and the Ratio is at maximum.

With these settings everything will be compressed very hard – even fast transients and impulses, resulting in an intense limitation of the output peak levels.

Now the Mix controller comes into play: If the just created ‚flat' signal is mixed to the original signal in moderate rates, it will be possible to achieve an increase in loudness while still saving a great part of the initial dynamics.

Stereo Enhancer

Even when there is nothing to compress from time to time, your productions can still benefit from the alpha compressor, as the combination of the MS matrix and the Audio filters makes a very nice tool to influence the stereo spectrum.

For this purpose the compressor is switched into MS mode and the Audio filter of the side channel adds 1,0-1,5dB to a center frequency of approx. 1kHz. The Gain controllers of both channels are at 0dB.

These settings cause a boost of all frequencies above 1kHz in the side channel, which intensifies the spatial perception as the room reflections become more distinct. The lower frequencies, however, are not of that great importance for spatial perception.

Of course simply increasing the level in the side channel can be used to achieve a (broadband) change in the stereo spectrum, too, but this variant offers much more differentiated options because it is frequency-dependent.

LoFi Compression

Can the alpha compressor also sound really bad? Of course not, but sometimes it just loves being, well … different.

In this LoFi Compression scenario the use of the Audio filters is the crucial point. By reducing the high frequencies from 4kHz on, everything sounds a little darker, while, at the same time, the bottom experiences a great degree of additional punch.

The chosen time parameters result in a loud and twangy control mode. The high Ratio settings create a high amount of compression, so that Gain reduction values of 8-12dB are absolutely normal.

In this application the Mix controllers give you the option to blend this extreme sound with the original at will, too.

Extreme Settings

Even though the main focus of the alpha compressor lies on mastering applications, you can also use it to find interesting and sometimes extreme settings for single channels or subgroups.

Here is just one of many examples. It is especially suitable for processing drum tracks effectively:

The time parameters with the additionally activated Auto Fast functions are very fast. The high Ratio in Feed Forward mode causes an extreme characteristic curve so that loud signals will be reduced significantly. By doing so, very high Gain reduction values of 20dB and more can occur by all means.

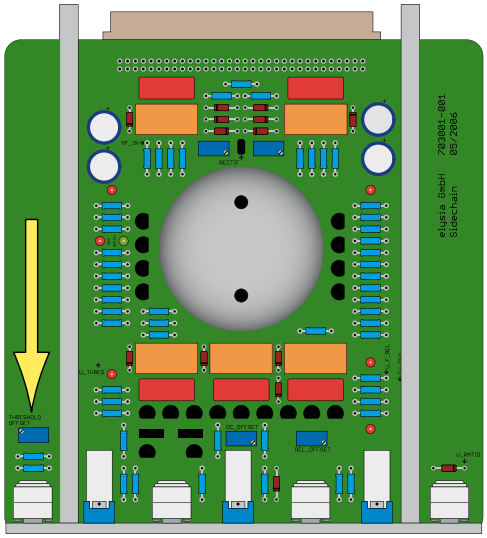
The spatial elements now come into the foreground more articulately – it somehow sounds as if the microphones had been placed a little bit further away during recording.

REFERENCE - Threshold Offset

The analog audio level of the alpha compressor depends on the studio environment it is used in. In the majority of cases it depends on the DA converter used, because the output stages of the converter determine the analog level. As there is no binding norm for it, this level can be at +4dB, +6dB, +10dBu, +12dBu or even +15dBu. To make sure that the Threshold controller actually covers its complete adjustment range it can be adjusted to those different levels.

When is an adjustment necessary?

1. At low Ratio values (1:1,3), the Threshold controller is almost completely turned clockwise and the Gain reduction is still too low
2. At higher Ratio values (1:2,5), the Threshold controller is turned completely counter-clockwise, but nevertheless Gain reduction already sets in

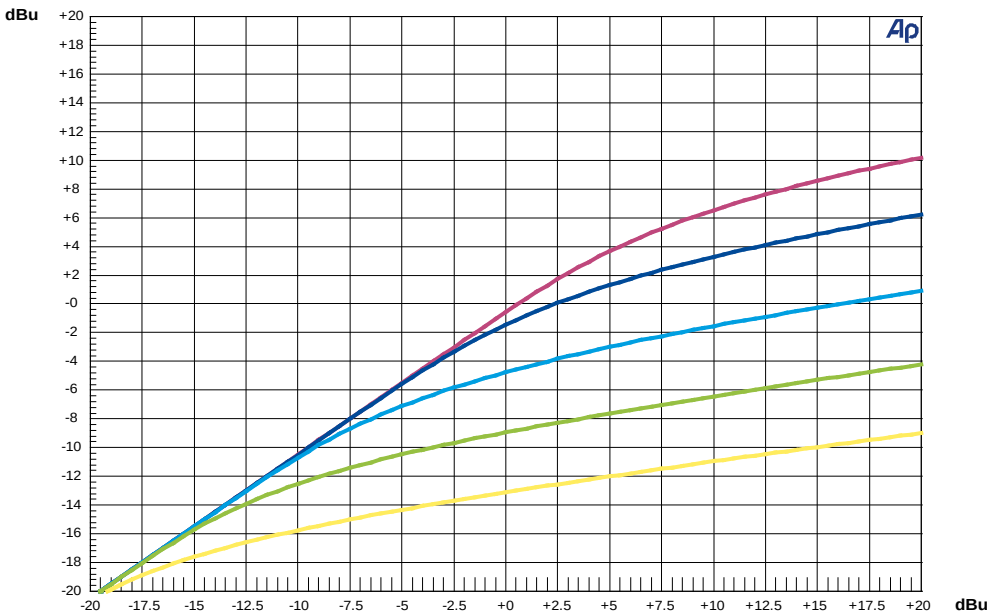


In order to adjust the internal Threshold Offset, the first thing to do is to open the top cover with the adequate Torx screwdriver. Before the adjustment can be done the unit has to be powered up for at least 15 minutes to reach a stable working temperature. Please make sure to disable the MS and Link functions. Now send an audio signal into the compressor and set the Attack to approx. 24ms and the Release to 300ms.

On the upper circuit board there is a blue trimmer labeled THRESHOLD OFFSET situated directly behind the Threshold potentiometer (the yellow arrow in the figure). In the first case (too little Gain reduction) the trimmer has to be turned clockwise to raise the amount of Gain reduction. In the second case (too much Gain reduction) the trimmer has to be turned counter-clockwise. In order to reassure that the adjustment range now really fits, it is recommended to repeat the above-mentioned test with the Threshold controller in extreme left/right position. The same procedure should also be applied to the right channel, of course.

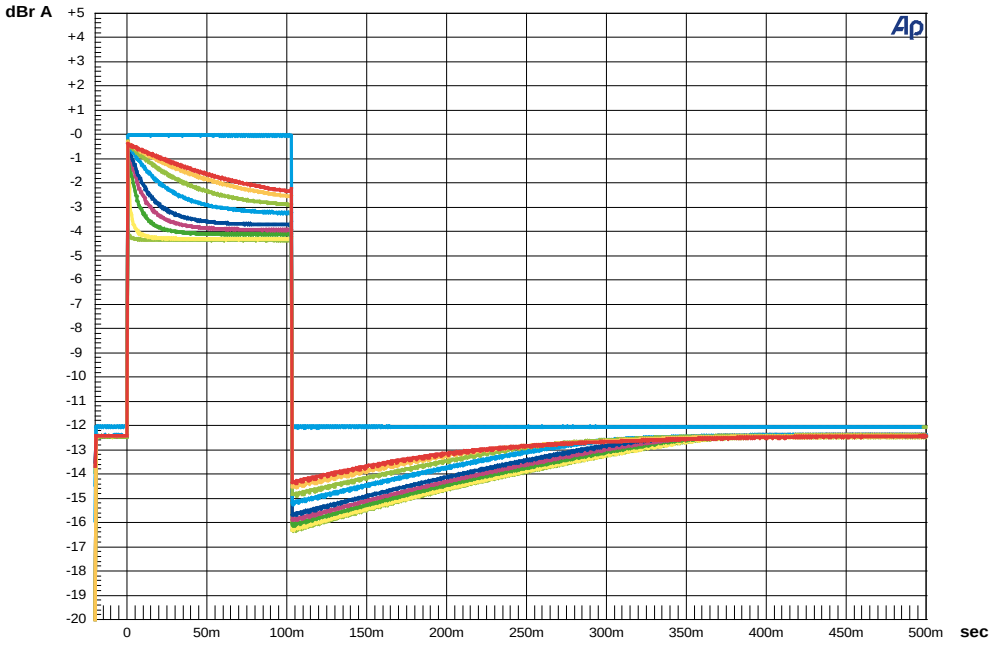
Threshold

The diagram shows different Threshold settings in Feedback mode. The Threshold and the Ratio are always interdependent: On high Ratios the Threshold will usually be set to lower values in order not to produce too much Gain reduction, and on lower Ratio settings the Threshold controller will be turned more to the right. The complete Threshold range covers 33dB which enables the user to make very sensitive settings in reasonable steps at any time.



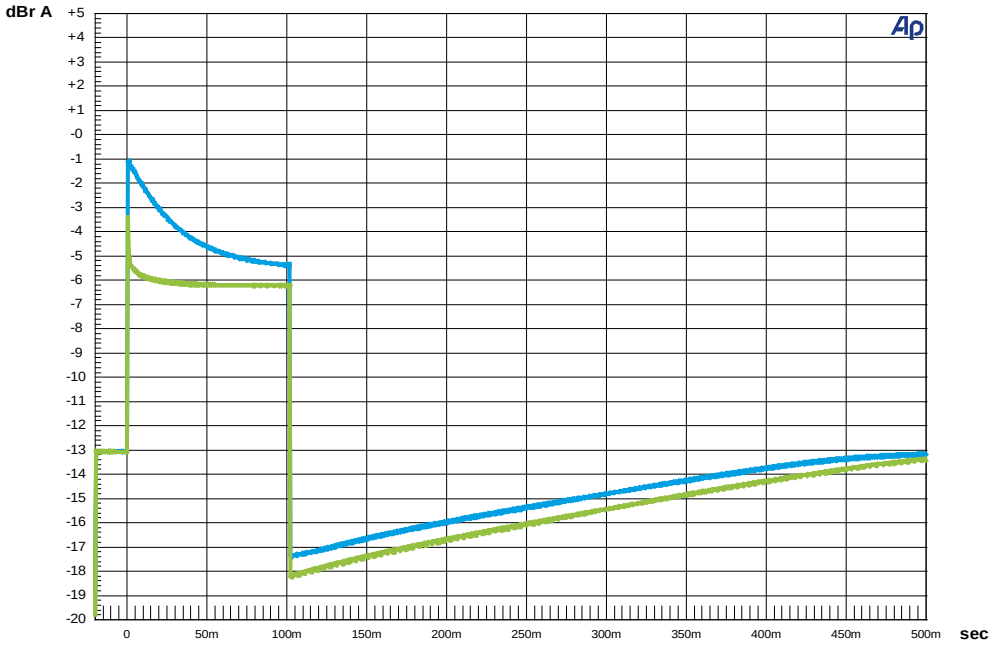
Attack

Here a Burst signal (a sine tone that is boosted by 12 dB for a period of 100ms) is used to demonstrate the effectiveness of the Attack controller. The different curves show how fast this level jump is reduced by 4dB. On fast settings the reduction takes only a few milliseconds, on longer settings the reduction does not only take longer but it is also smaller. Therefore on fast Attack settings the Gain reduction is always higher than on longer settings.



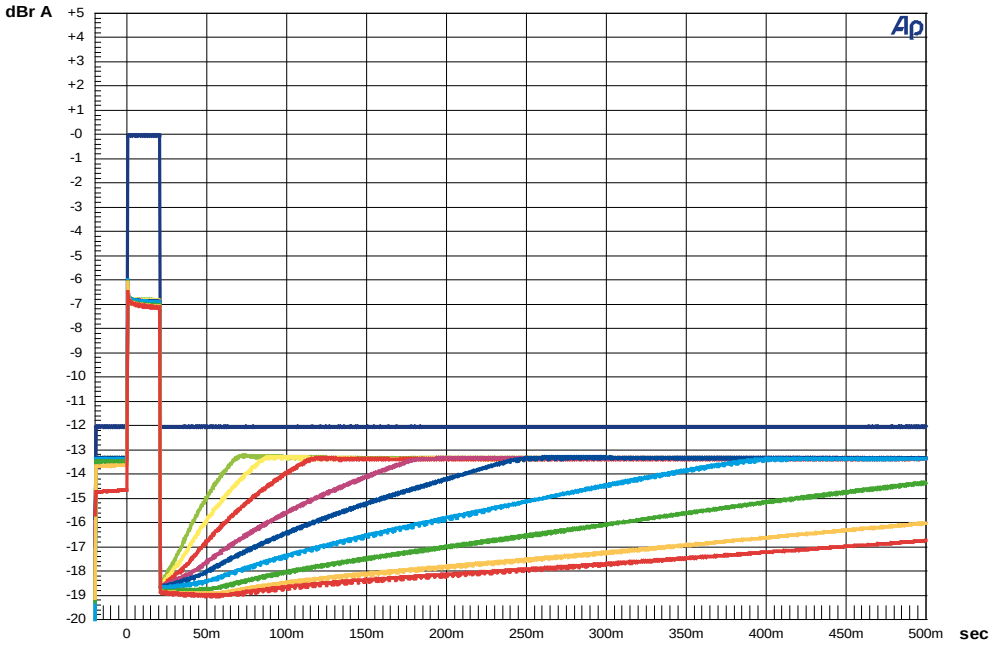
Auto Fast Attack

Utilizing a Burst signal again, this diagram shows a comparison between a normal Attack time of 50ms (blue curve) and the later added Auto Fast function (green curve). This reveals how fast this circuit reacts: Even the first waves of the Burst are already reduced by 2,3dB. Also the Gain reduction as a whole now happens much faster. *Note*: This function starts working at -3dB; on lower Gain reductions the changes in dynamics are too small.



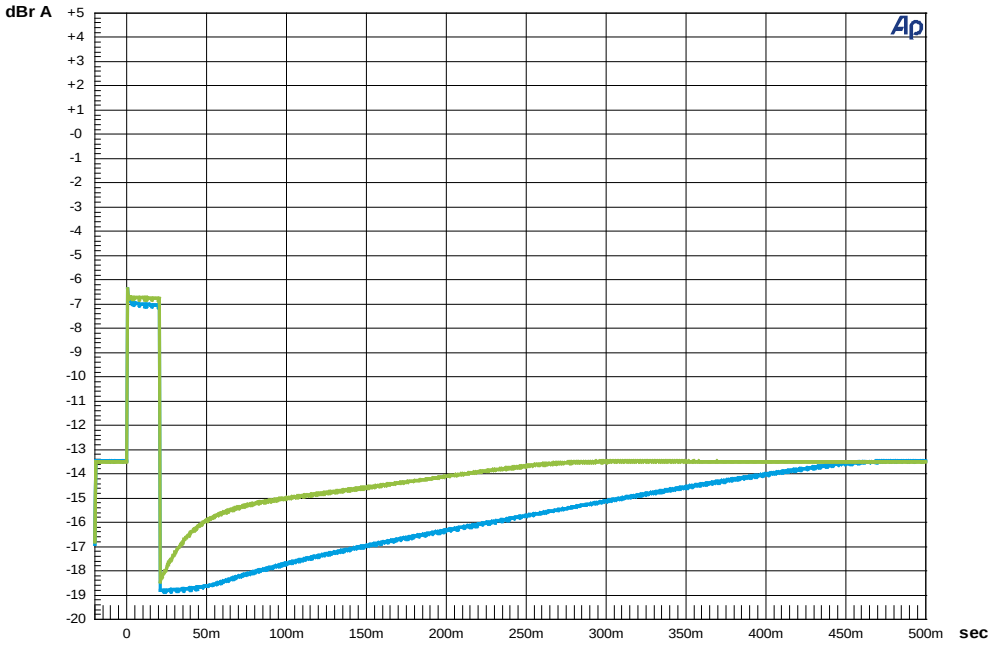
Release

This figure is based on a very fast Attack time; the Gain reduction is at -7dB. The Burst lasts for 25ms and then follows the Release phase with time parameters of different length. The linear characteristic of the Release curve is very easy to see here. Because of the utilized circuit the Attack time must always be added to the Release time. The advantage is that in combination with long Attack settings you can use even the fastest Release settings without generating unwanted artifacts.



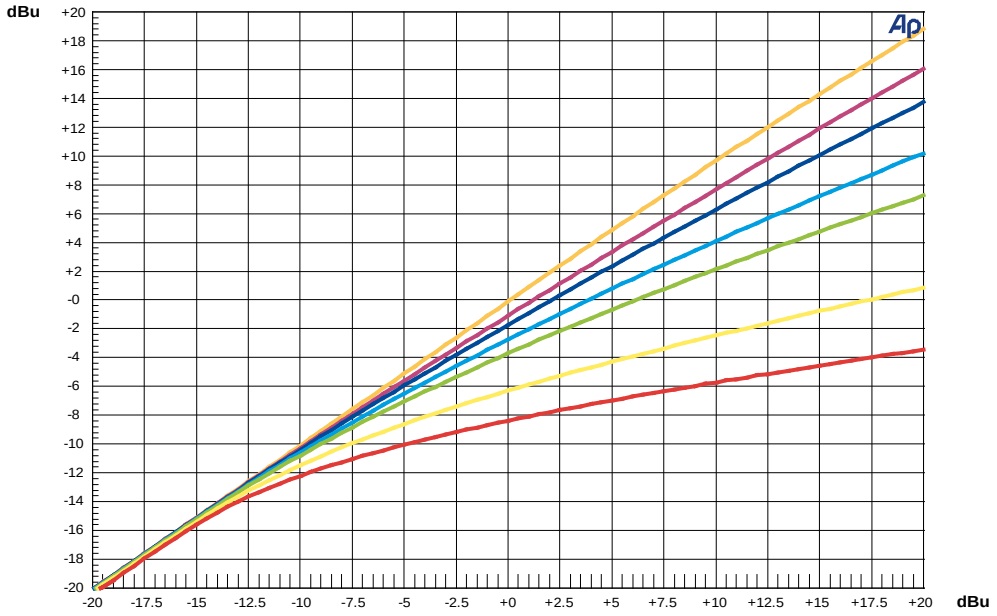
Auto Fast Release

In this case the Auto Fast function was activated at a Release time of 400ms (green curve). It is easy to see that directly after the Burst the Release time becomes very fast (only 20ms) and then returns to its original setting. This has two effects: firstly, the level in a period of 500ms after the Burst becomes 2,5dB louder and secondly the Release phase is already completed after 250ms. In practice this means an apparent increase in loudness without the distortion usually generated by constantly fast Release times.



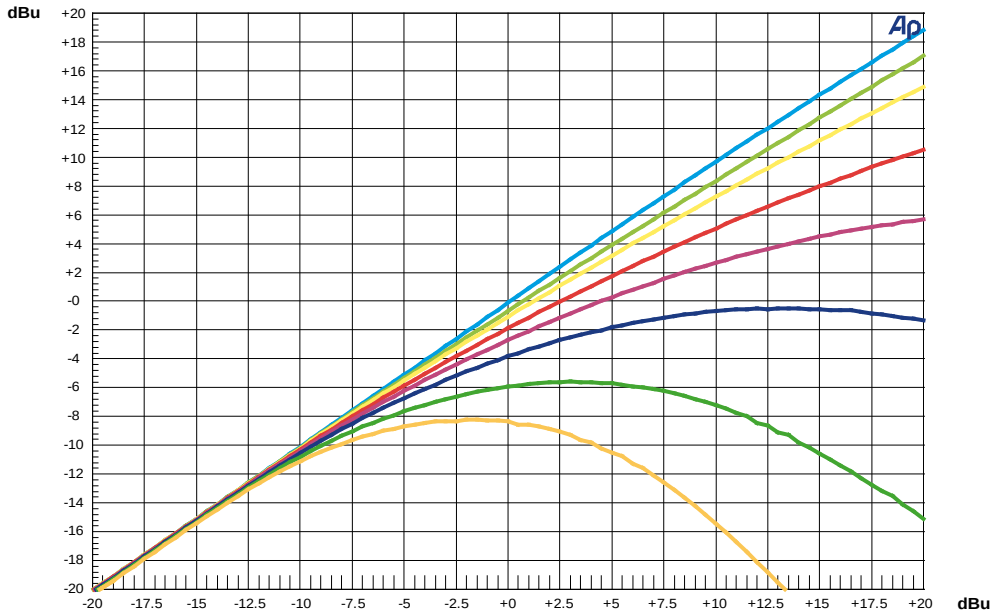
Ratio Feedback

The alpha compressor works with a soft knee characteristic and increasing Ratios at rising Gain reduction values. The scale on the front panel is based on a measured Gain reduction of 6dB. Strictly speaking, the Ratio values become noticeably higher on stronger reductions: When there is 17dB of Gain reduction the ratio value shifts from the printed 1:2,5 to 1:4,5 after all. Anyhow, the scale is optimized to the lower Gain reduction settings as these are the ones most frequently used in mastering applications.



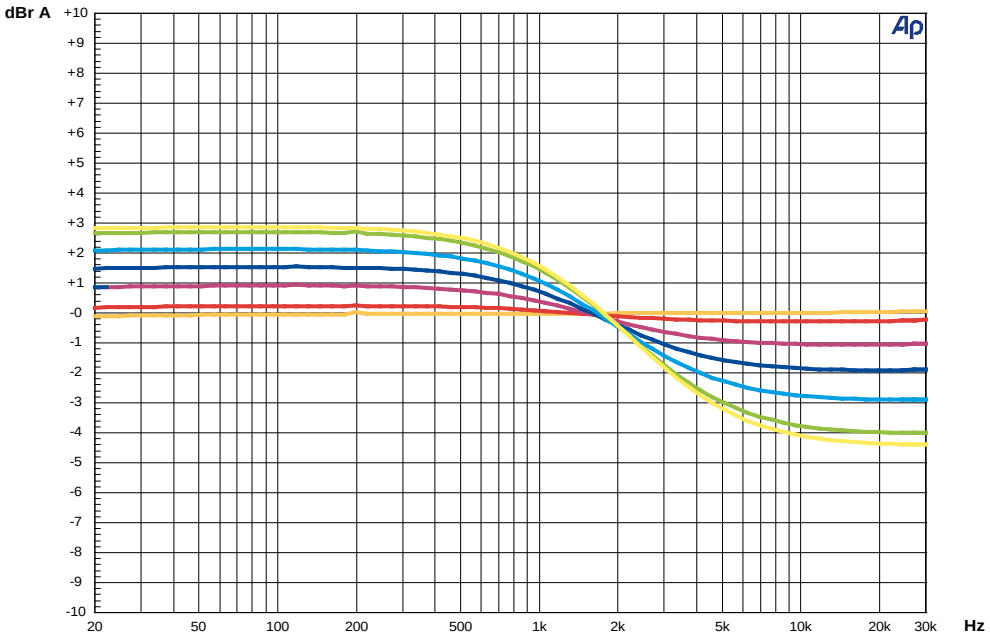
Ratio Feed Forward

In Feed Forward mode the Ratio characteristics differ strongly from those in Feedback mode. Although they remain quite comparable up to a value of 1:1,5, the differences become extremely obvious at higher settings. The lower curve shows the radical effects: An input level of 0dBu is reduced to -8,3dBu, and an input level of +10dBu is reduced to -15dBu (!) after processing. Extreme settings like these can only be achieved with a Feed Forward compressor, and they can result in very freaky effects.



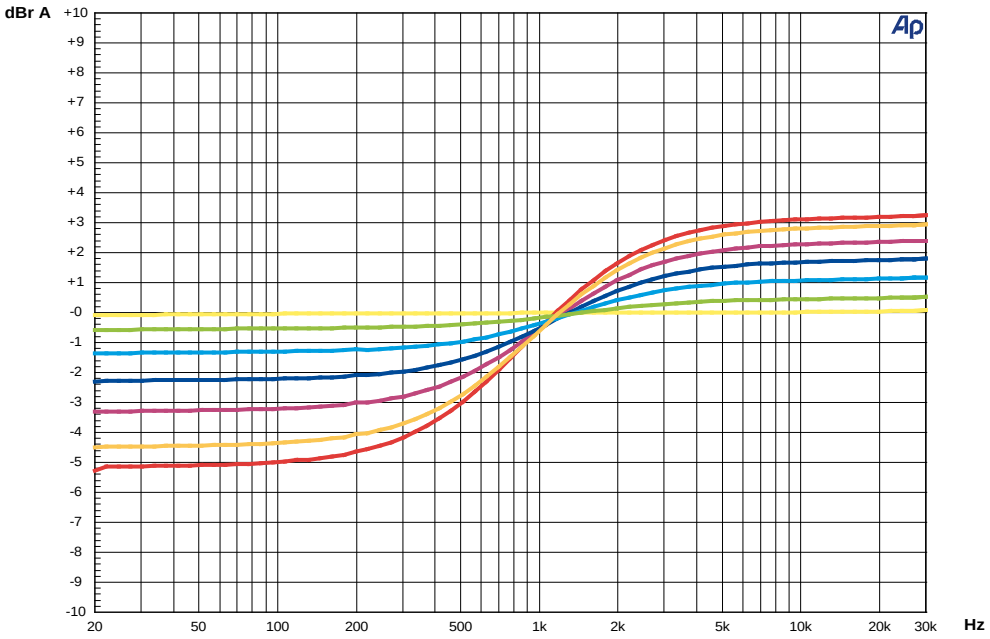
EQ Gain Low

For this measurement the center frequency was set to 1,8kHz and the EQ Gain controller was set to different positions between 0 and full left (-3dB). The diagram shows the effectiveness of the filter: The lower frequencies are boosted up to 3dB, at the center frequency they cross zero, and the higher frequencies are cut up to 4,5dB. Depending on the setting of the EQ Gain the level might have to be adapted with the Gain controller.



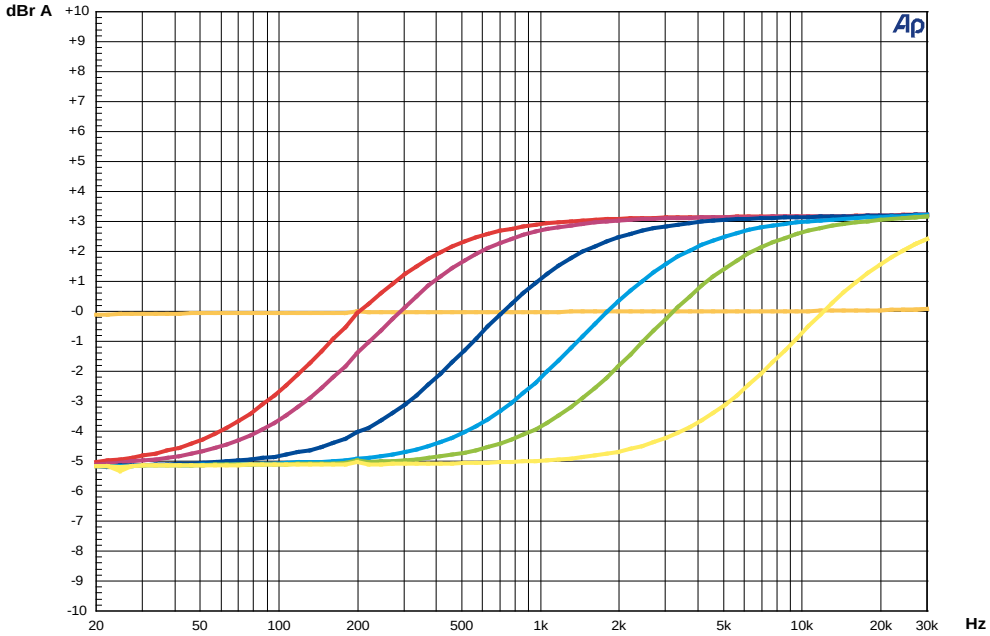
EQ Gain High

Now let's have a look at the opposite scenario. In this figure the EQ Gain controller was turned from 0 to full right at a center frequency of 1,1kHz. Contrary to the previous example the high frequencies are now boosted up to 3dB and the lower frequencies are reduced by up to 5dB. The fine-tuned interim values make it possible to apply very subtle sonic changes.



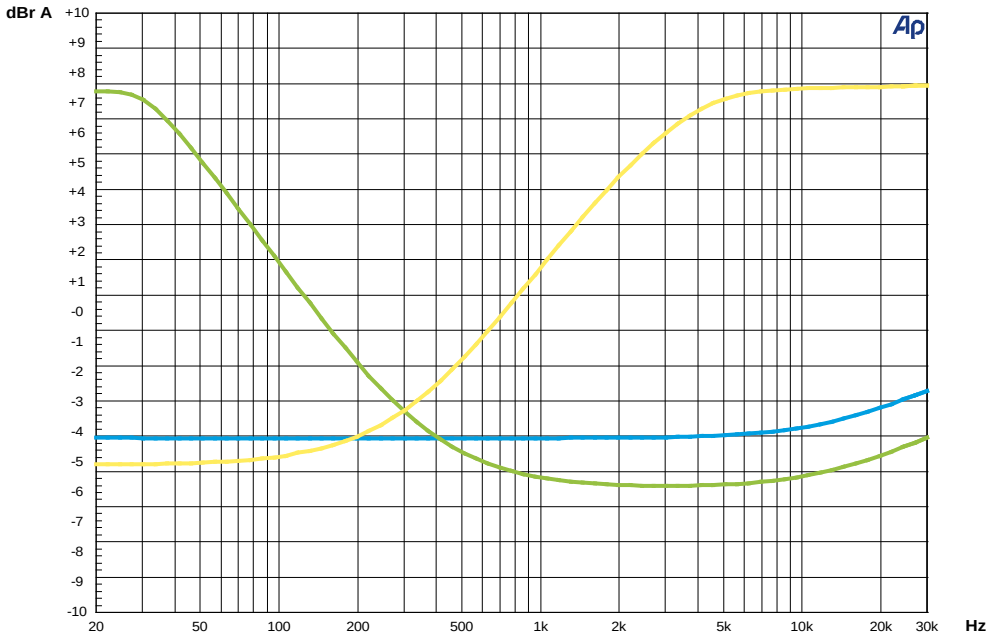
EQ Frequency

This diagram shows a variation of center frequencies with the EQ Gain controller fully turned clockwise (HI). The x10 button shifts the complete frequency area by a factor of 10. With settings in the border areas of the covered frequencies it is also possible to realize "almost" shelving filters, but in these cases the level has to be adapted with the Gain controllers.



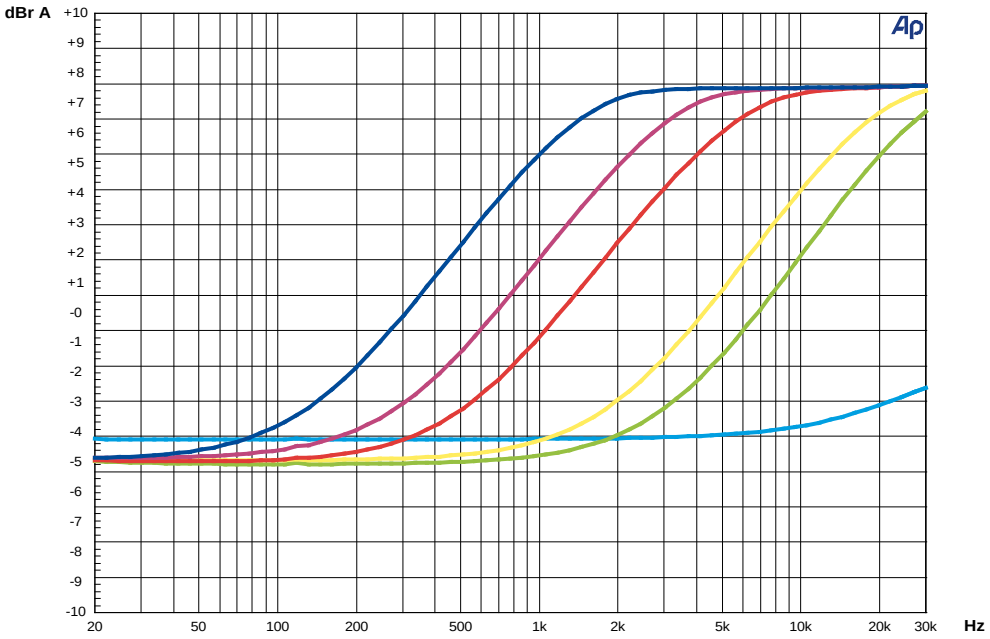
SC Gain

The blue curve shows an almost linear compression with a constant Gain reduction across the complete frequency spectrum. If the SC Gain controller is set to High Pass (HP – green curve), however, the low frequencies will not influence the detection circuit and the Gain reduction sets in only on higher frequencies. In Low Pass mode (LP – yellow curve) just the low frequencies are fully processed and the intensity of the Gain reduction decreases with rising frequencies. Of course there are a lot of useful interim values between the linear and the extreme positions.



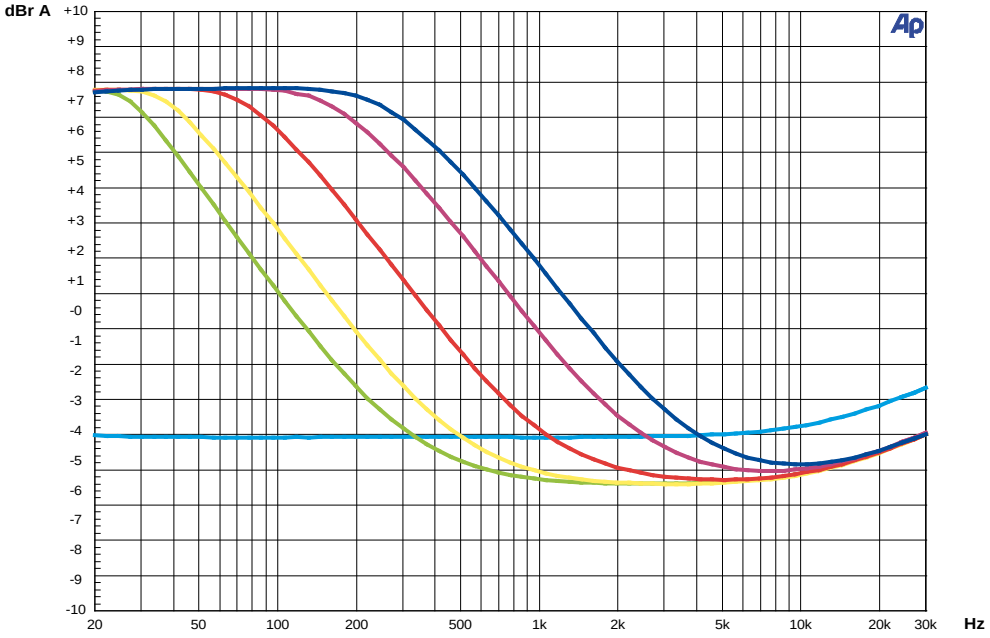
SC Low Pass

This diagram shows some settings of the Sidechain frequency controller (SC Freq) in Low Pass mode (LP). If the filter is set at 100Hz (dark blue curve) for example, full Gain reduction will still occur at these 100Hz and the high frequencies will have lesser influence the higher they get. Therefore in this mode of operation the filter should always be set to the highest frequency you just about want the compressor to react to.



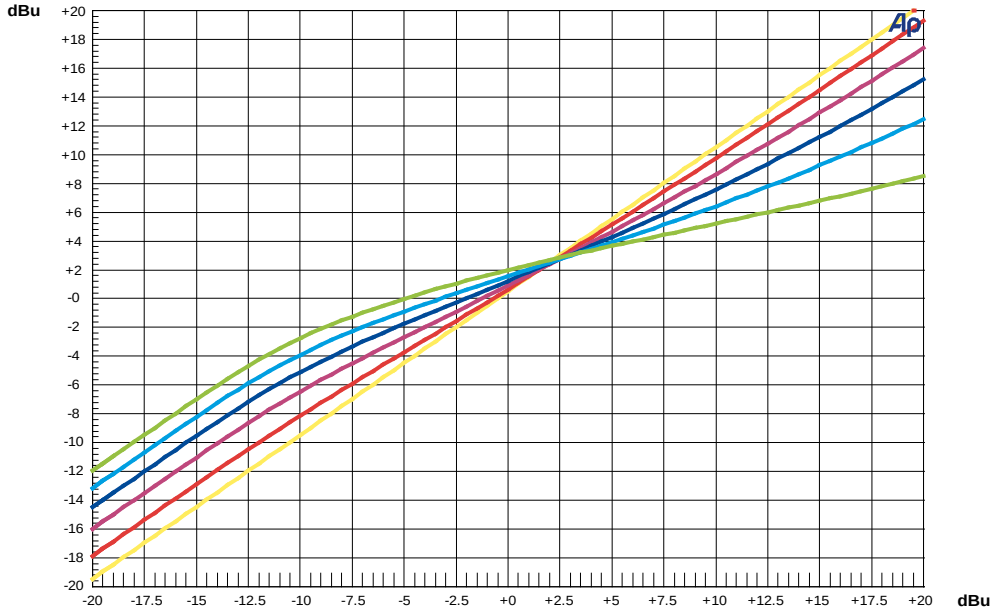
SC High Pass

Here the controller was turned completely the other way round. In High Pass position (HP) the approach of the Sidechain filter is slightly different. At 100Hz the compressor now hardly reacts to just this frequency and only the higher frequencies will have an increasing influence on the Gain reduction. In practice the controller should therefore be set to the lowest frequency you just about want the compressor not to react to.



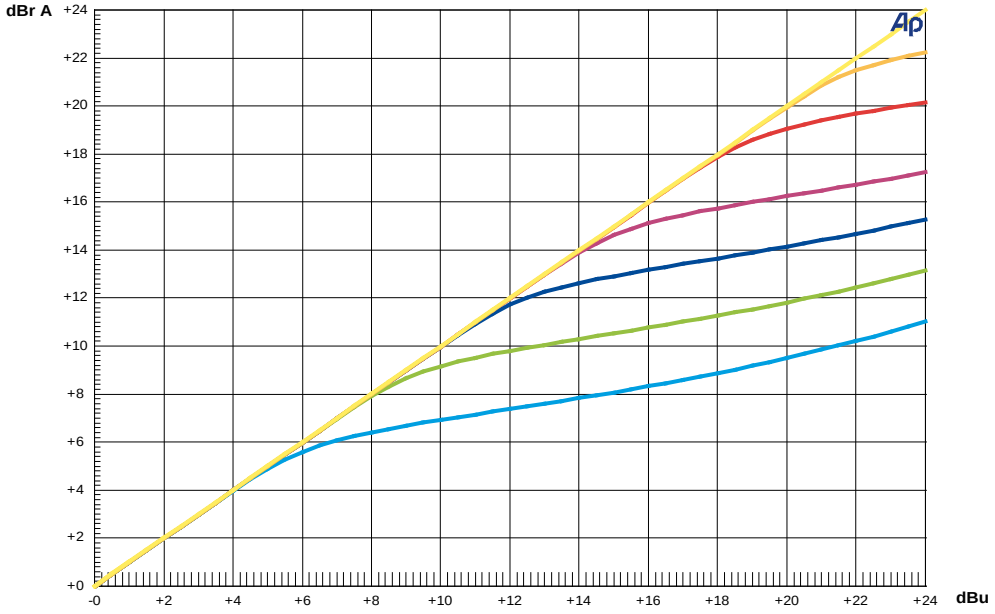
Mix

This measurement diagram shows a fade from the original to the compressed signal (Direct and Compressed buttons active). The Gain controller was set at 0dB, which is also the reason for decreasing levels in the more clockwise areas of the controller range. In case that only the direct or alternatively only the compressed signal is switched in, the Mix controller does not have any influence on the level.



Soft Clip

These characteristic curves show the smooth processing of the Soft Clip limiter. Even though this figure shows huge amounts of level reduction to demonstrate the effect clearly, the limiter values should not exceed 6dB as a general rule. After all, this function aims at rounding the peaks of a signal and not at reducing the complete level. Too heavy use of the Soft Clip limiter can cause audible distortion.



APPENDIX - Level Problems

No Signal

This problem might be caused by balanced wiring. If the signal almost completely drops when the alpha compressor is activated, it is probable that a pin of the XLR connector at the input is not connected. As the input stages feature transformers, both pins have to be connected. The classic case for this problem would be a balanced XLR connection that is connected to an unbalanced output that only uses ground and pin 2. Connecting pin 3 to ground should solve this problem.

Slight Level Jump

Under certain circumstances, pushing the active switch may cause a small change of level. If there is no noticeable difference between Bypass and Active when in stereo mode with activated Direct buttons and Mix controllers fully turned counter-clockwise, everything is OK. But if there is a small change of level somewhere between 0,5 - 1dB, the input and output impedances of the devices preceding and succeeding the alpha compressor might be the reason. A high output impedance (approx. 1kOhm) of the preceding device and a low input impedance (<4700Ohm) of the succeeding unit are especially critical. If the compressor is in Bypass, the preceding device will be loaded with the input impedance of the alpha compressor itself plus the input impedance of the succeeding unit. This cumulative load might result in a decrease of level. If now the alpha compressor is activated, the preceding device only "sees" the alpha compressor and no longer the following unit. Therefore the load decreases and the level rises as a consequence. Especially tube equipment tends to produce this phenomenon. Generally speaking, output impedances smaller than 150Ohm are recommended for a trouble-free signal chain.

Large Level Jump

In some equipment the output stage is designed in a way that the level will always stay the same – no matter if it is hooked up with a balanced or an unbalanced connection. If pin 3 is connected to ground, for example, the level at pin 2 will automatically become twice as loud as before. These kinds of output stages are usually unproblematic. But there are also stages that cannot compensate for that. In these cases the level at pin 2 will stay as it is, even if pin 3 is connected to ground. If the alpha compressor is placed between a device with that kind of output stages and a device with unbalanced inputs and with pin 3 connected to ground, it is possible that the level jumps up by 6dB when the compressor is activated. As a general rule, input stages with balanced connectors are always the best choice. In case these are not available, the first attempt to solve this problem should be to disconnect pin 3 at the XLR inputs of the alpha compressor and then connect it to ground. This generates an unbalanced signal that should not shift the level anymore.

Technical Data

Frequency Response:	<10Hz – 200kHz (-0.5dB)
THD+N @+15dBu, 20Hz-22kHz:	
Stereo Mode (Direct)	0.0039%
Stereo Mode (Compressed)	0.009%
MS Mode (Direct)	0.014%
MS Mode (Compressed)	0.034%
Noise Floor, 20Hz-20kHz (A-weighted):	
Stereo Mode (Direct)	-95.8dBu
Stereo Mode (Compressed)	-89.3dBu
MS Mode (Direct)	-95.6dBu
MS Mode (Compressed)	-92.3dBu
Dynamic Range, 20Hz-22kHz:	
Stereo Mode	122dB
MS Mode	118dB
Maximum Input Level:	
Stereo Mode	+28dBu
MS Mode	+23dBu
Maximum Output Level:	
Stereo Mode	+27dBu
MS Mode	+23dBu
Input Impedance:	10kOhm
Output Impedance:	68Ohm
Pin Assignment Input:	1. GND 2. Positive (transformer balanced) 3. Negative (transformer balanced)
Pin Assignment Output:	1. GND 2. Positive 3. With 68Ohm to GND
Power Consumption:	100W
Fuse Type:	115VAC 3A Slo-Blo / 230VAC 1,5A Slo-Blo
Dimensions (W x H x D):	483mm x 133mm (3U) x 405mm 19“ x 5.3“ (3U) x 16“
Weight:	16kg / 35,27lb

Warranty

Conditions and limitations

The alpha compressor is covered by a limited warranty for a period of 5 years against defects in parts and labor from the date of purchase. Natural wear is not covered by this warranty. elysia will remedy problems caused by material or workmanship either by repair or replacement to restore the product to full performance without charge for parts and labor. Repairs or replacements will not extend the warranty period.

The warranty is given to the original purchaser only and is not transferable. elysia will only give warranty on products purchased through authorized elysia dealers. The warranty will only be valid in the country of the original purchase unless otherwise pre-authorized by elysia.

All warranties become void when the product has been damaged by misuse, accident, neglect, modification, tampering or unauthorized alteration by anyone other than elysia authorized service personnel.

The warrantor assumes no liability for property damage or any other incidental or consequential damage whatsoever which may result from failure of this product. Any and all warrantees of merchantability and fitness implied by law are limited to the duration of the expressed warranty.

This warranty gives you specific legal rights and you may also have other rights which vary from state to state. Some of the above limitations may not apply to you.

Registration and return

To confirm your warranty please register your elysia product shortly after purchase. The easiest and fastest way is to use the online form you will find in the Service section of our website www.elysia.com. If you cannot or do not want to use this option, please contact us via the address you'll find at the end of this warranty statement. You will be provided with registration documents suitable for fax or mail use free of charge.

In case you notice any defect, please contact elysia for technical support directly. You can find the correspondent contact data at the end of this warranty statement. You will receive a return authorization which enables you to send your product to the elysia factory where it will be repaired and then sent back to you.

Packaging and shipping

All returns to the factory must be in the original packaging, accompanied by the return authorization, and must be shipped via insured freight at the customer's own expense. A new original packaging can be ordered from elysia. The customer will be charged for new factory original packaging if he fails to ship the product in the original factory packaging.

In case that a product must be returned to the factory from a country outside Germany, the customer shall adhere to specific shipping, customs, and commercial invoicing instructions given with the return authorization as elysia will not be responsible for transportation costs or customs fees related to any importation or re-exportation charges whatsoever.

After repair, the product will then be returned to the customer via prepaid, insured freight, method and carrier to be determined by elysia. elysia will not pay for express or overnight freight service or pay for shipments to locations outside Germany. All damages caused by transport are not covered by this warranty.

Contact data

For technical support and return authorizations, please contact:

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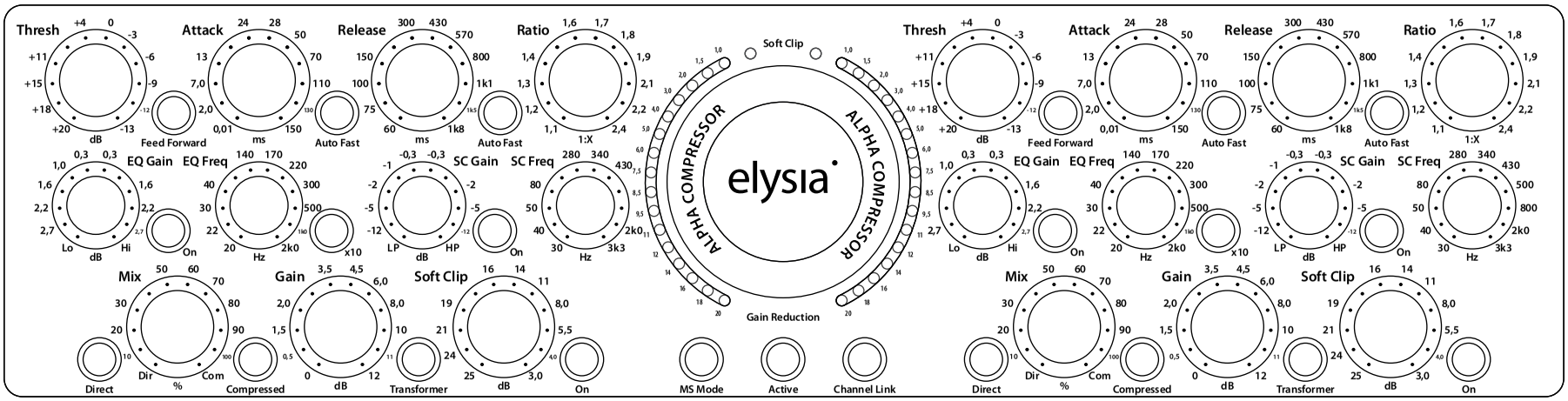
D-41307 Nettetal
Germany
Tel: +49 (0) 21 57 / 12 60 40
Fax: +49 (0) 21 57 / 12 63 12
service@elysia.com



Project . . . . . Song . . . . .

Engineer . . . . . Date . . . . .

Left Channel Track . . . . . Right Channel Track . . . . .

Notes . . . . .

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„Let there be sound“

And there was sound.

(AC/DC)